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# Performance of TCP Variants over LTE Network

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**Abstract.** One of the implementation of a wireless network is based on mobile broadband technology Long Term Evolution (LTE). LTE offers a variety of advantages, especially in terms of access speed, capacity, architectural simplicity and ease of implementation, as well as the breadth of choice of the type of user equipment (UE) that can establish the access. The majority of the Internet connections in the world happen using the TCP (Transmission Control Protocol) due to the TCP's reliability in transmitting packets in the network. TCP reliability lies in the ability to control the congestion. TCP was originally designed for wired media, but LTE connected through a wireless medium that is not stable in comparison to wired media. A wide variety of TCP has been made to produce a better performance than its predecessor. In this study, we simulate the performance provided by the TCP NewReno and TCP Vegas based on simulation using network simulator version 2 (ns2). The TCP performance is analyzed in terms of throughput, packet loss and end-to-end delay. In comparing the performance of TCP NewReno and TCP Vegas, the simulation result shows that the throughput of TCP NewReno is slightly higher than TCP Vegas, while TCP Vegas gives significantly better end-to-end delay and packet loss. The analysis of throughput, packet loss and end-to-end delay are made to evaluate the simulation.

## INTRODUCTION

Long Term Evolution (LTE) is an evolution of mobile network technology in the world that offers variety of advantages, especially in terms of access speed, which provides a level of capacity of at least 100 Mbps downlink and uplink at least 50 Mbps and round-trip time (RTT) of less than 10 ms [1]. In a telecommunication system network, the inevitable exchange of data involves data exchange protocols in the Open Systems Interconnection (OSI). One of the layer in OSI is the transport layer. In addition, Transmission Control Protocol (TCP) is a protocol that works at the transport layer of the OSI model (Layer 4) which serves to transmit data per segment, meaning that data packets are sent in burst in an amount corresponding to the amount of the package then sent one by one to finish. In order for data transmission up well, then at the time of each packet transmission, TCP will include a serial number as known as sequence number. LTE supports deployment on different frequency bandwidths. The current specification outlines the following bandwidth blocks: 1.4MHz, 3MHz, 5MHz, 10MHz, 15MHz, and 20MHz [2].

The growing popularity of LTE networks has led to cases of heavy utilization and congestion. Network congestion is a phenomenon in which the burden exceeds the capacity of the network [3]. On the other hand in the development of data networks, TCP is a protocol that works at the transport layer of the OSI model and pretty much used, as a connection oriented protocol that has congestion avoidance mechanisms to ensure the delivery of data packets are not lost and get to the destination.

The most popular version TCP and most widely used today is TCP NewReno. However, the TCP NewReno is considered less effective in terms of media utility when congestion in the network and when should pass in certain medium such as wireless. Therefore, there are many algorithms have been proposed and implemented to improve the performance of TCP, such as TCP Vegas.

In this section, we provide an overview of congestion and TCP congestion control, especially those closely related to our work.

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## Congestion

Definition of congestion is as follows: A network congestion from a user perspective if the quality of service perceived by the user decreases as the increase in network load. If the time allocation for each user has reached the minimum threshold, but the load is still increasing, the allocation will be smaller. If this is the case, the allocation will reach a value small enough such that the perceived user cannot perform data communication. This condition is called congestion [4].

## TCP Congestion Control

At today's TCP, the core of congestion control is to adjust the variable congestion window (cwnd), which determines how many packages are not recognized by sender can be sent. Congestion control algorithms which differ primarily determine how the congestion window should be increased for each incoming ACK (acknowledgement) packet and how the congestion window should be decreased to every event of congestion. TCP congestion control was first proposed by Jacobson as a means to prevent "congestion collapse", a condition in which too much traffic on the network led to excessive packet loss [5].

TCP NewReno is most widely studied as the basic congestion control algorithm, which is the base algorithm implemented in the Linux TCP stack. It uses the traditional additive-increase, multiplicative-decrease (AIMD) to control the cwnd. In other hands, NewReno increases the cwnd linearly by one packet for every round-trip time and decreases it by half for every congestion event. One good advantage of AIMD algorithm is that it allowed the cwnd of multiple flow through a link to converge to a fair value [6].

TCP Vegas was the first algorithm that proposed using packet delay or RTT over packet loss as the main signal for congestion. It records the minimum RTT value and uses it to calculate an expected rate. The expected rate is then compared with the actual rate and the cwnd is additively increased, kept constant, or additively decreased [7]. Several studies establish that TCP Vegas does achieve higher efficiency than NewReno, causes the end-to-end delay in TCP Vegas less than NewReno, and it is not biased against the connections with longer RTTs [8].

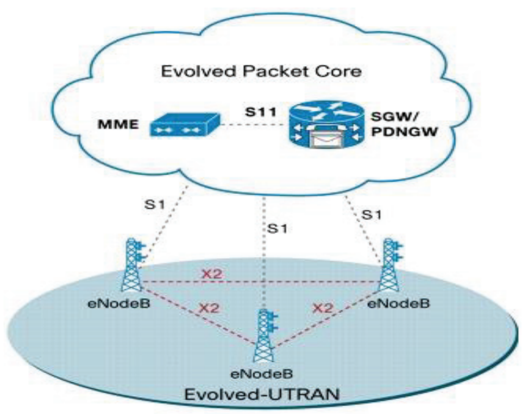


**FIGURE 1.** LTE representation

## EXPERIMENT METHOD

A simple LTE architecture [9] that has been shown in Figure 1 consists of one server for serving FTP (File Transfer Protocol) and provide source connection for the TCP link over the topology. In LTE system, the main job of aGW router is to control the flow rate of the streaming data from server to user equipment (UE) called evolved-NodeB (eNB), where these nodes responsible for buffering the data packets for UE over the network. Each eNB is connected to the corresponding aGW through wireless of 11 MHz bandwidth. The proposed topology has shown two UEs are used, and connected to eNB within constant bandwidth 11 MHz. In this study, the type of TCP which used are TCP NewReno and TCP Vegas and for the traffic used is FTP. The goal of our experiments is to understand the performance of TCP NewReno and TCP Vegas over a network topology based on LTE system. We evaluated performance of the proposed model by using NS-2 simulator [9]. NS2 network simulator simulates network based on TCP/IP with a wide variety of media we can simulate network protocols such as TCP, UDP (User Datagram Protocol), RTP (Real-time transport protocol), traffic behavior like FTP, Telnet, CBR (Constant Bit Rate), Queue Management, Unicast Routing Algorithm, Link State, Multimedia Applications such as video layer, quality of service, and audio-video transcoding. The parameters of modeling and simulation are presented in the following table [10].

**TABLE 1.** Simulation Parameters

| Parameter | Value |
|---|---|
| TCP Protocols | TCP NewReno and TCP Vegas |
| Bandwidth | 11 MHz |
| Propagation Model | Two Ray Ground |
| Packet Size | 1500 Bytes |
| Simulation time | 50 Seconds |
| Traffic | FTP |

## RESULT AND ANALYSIS

In the simulation model shown in Figure 1, the nodes 0 to 5 are established with same parameters and behavior. Figures 2 and 3 represent the comparison of NewReno and Vegas under similar network conditions, where the bandwidth, propagation model, packet size, simulation time, and the traffic are kept the same as Table 1.

Figure 2 shows the relationship packet delivery and increasing throughput occurs at the beginning of sending packet and relatively stable when it is sending packets, The simulation result shows that the throughput of TCP NewReno is slightly higher than TCP Vegas. The average throughput value of TCP NewReno is 1033.34 kbps and the average throughput value of TCP Vegas is 912.624 kbps.

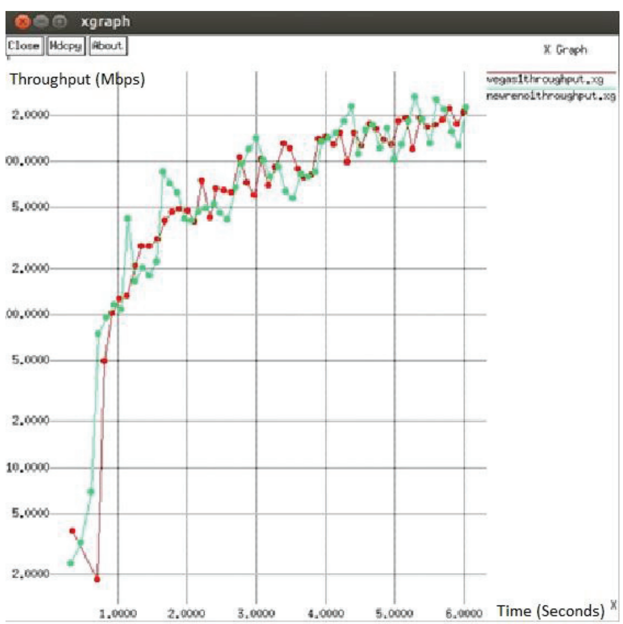


**FIGURE 2.** TCP NewReno and TCP Vegas throughput

**TABLE 2.** Packet loss and End-to-End Delay

| TCP | Packet Loss | End-to-End Delay (Average) |
|---|---|---|
| TCP NewReno | 10.07 % | 0.46724 |
| TCP Vegas | 8.75 % | 0.13720 |

In the Table 2 above, we compare the packet loss on TCP NewReno and TCP Vegas and the end-to-end delay occurs, we found the packet loss occurs in TCP NewReno is higher than TCP Vegas and the end-to-end delay measured on TCP NewReno and TCP Vegas shows that TCP Vegas has less end-to-end delay than TCP NewReno. It means that TCP Vegas is better than TCP NewReno in the process of data delivery in terms of packet loss and end-to-end delay.

## CONCLUSION

As a conclusion, it is shown that the performance of TCP Vegas is better than TCP NewReno in sending FTP data in terms of throughput, packet loss and end-to-end delay over LTE network.

TCP Vegas gives better packet loss, end-to-end delay than TCP NewReno, whereas TCP NewReno has slightly higher throughput than TCP Vegas. The result conforms to the studies conducted by other researchers [7, 8] regarding the performance of TCP NewReno and TCP Vegas over LTE network for the FTP data delivery. For this reason, TCP Vegas is recommended to be used as a transport protocol for LTE Network.

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